

10/518208
DT01 Rec'd PCT/PTO 16 DEC 2004**APPARATUS AND METHOD FOR RESOURCE ALLOCATION****TECHNICAL FIELD OF THE INVENTION**

The present invention relates to the handling of link and cell congestion in packet transmission networks and more particularly to the early detection of congestion and the implementation of mechanisms for obviating the consequences of congestion.

DESCRIPTION OF RELATED ART

In packet based communication systems, i.e. in which information to be transmitted is divided into a plurality of packets and the individual packets are sent over a communication network, variable bit rates occur. It is therefore known to provide queue buffers at various points in the network to accommodate for sudden bursts in the load.

A phenomenon that is known in packet transmission networks is that of link congestion. Link congestion implies a state in which it is not possible to readily handle the number of data packets that are required to be transported over that connection or link. As a consequence of congestion at a given link, the number of data packets in a queue buffer associated with said link will increase and buffer overload will occur. In response to a link congestion condition, it is known to implement a data packet dropping mechanism referred to as "drop-on-full". According to this mechanism, upon receipt of a new data packet at the queue buffer, a queue length related parameter, such as the actual queue length or the average queue length, is compared to a predetermined threshold. If the predetermined threshold is exceeded, then a data packet is dropped. The threshold indicates the "full" state of the queue.

The so-called "Transmission Control Protocol" (TCP) is a commonly used protocol for controlling the transmission of packets over an IP network. When a TCP connection between peer hosts is initiated, TCP starts transmitting data packets at a relatively low rate, i.e. so called "slow start mode". The transmission rate is successively increased in response to receipt of acknowledgement from the receiver. If data packets are detected as missing, then TCP interprets this as an indication of congestion and reduces its load.

Compared to wired networks, wireless links are equipped with a rather limited capacity. This is why it can be expected that the wireless link will often be the bottleneck of an end-to-end connection. This means that excessive load of a TCP connection will eventually build up in the buffer prior to the congested link. Since the buffer contributes to the end-to-end delay, it is desirable to keep the buffer as small as possible since large delays cause sluggishness to interactive traffic. At the same time, however, the buffer should be large enough to smooth out load variations, in order to utilise the capacity allocated for the link.

Further, the dynamics of TCP is strongly dependent how or in which order segments are discarded. Consecutive segment losses are likely to put the connection into TCP slow start, which is particularly bad for high-latency links, such as wireless links.

To fulfil these requirements on the buffer, Active Queue Management (AQM) may be used. The principle of Active Queue Management is to detect congestion at an early stage, before the buffer overflows. When congestion or near congestion is detected, it is alleviated by e.g. discarding packets or signalling congestion using Explicit Congestion Notification (ECN) according to some given Active Queue

Management algorithm. Typically, an algorithm is used for indicating congestion, without discarding all incoming packets.

Random Early Detection (RED) - see e.g. Floyd, S. and
5 Jacobson, V. "Random Early Detection Gateways for
Congestion Avoidance", IEEE/ACM Transactions on Networking,
1(4), August 1993 - is an Active Queue Management method
that has found wide acceptance within Internet Routing. The
10 RED principle is that an incoming packet is accepted if the
queue level is less than a low fixed queue threshold, but
discarded if the queue level is greater than a high fixed
queue threshold. For intermediate queue fill levels,
incoming packets are discarded with a certain probability.

Other solutions for Active Queue Management algorithms are
15 also described in EP 01107850.8 (filed, but not published
at the filing date of the present application) and
GB 0113214.1 (filed, but not published at the filing date
of the present application).

In systems with limited resources, congestion may also
20 occur in a larger concept than in the individual link and
some sort of resource management may therefore be employed.
This is especially the case in mobile networks.

A mobile network includes among other things a set of base
stations or node Bs, each serving a given cell or a number
25 of cells. A mobile station or user equipment may connect to
one or more base stations to make or receive a call. If the
mobile station moves from one cell to another during a
call, handover may occur, meaning that the mobile station
now communicates with another cell and possibly another
30 base station. Different types of handover exist.

A link is in this context a service provided for
transmission of data packets between a mobile network and a
mobile station or user equipment. Communication from the

mobile network to the mobile station or user equipment is referred to as a downlink, while communication from the mobile station or user equipment to the mobile network is referred to as an uplink.

5 The code division multiple access (CDMA) communication method was developed to allow multiple users to share radio communication resources. In the general CDMA method, each user is assigned a unique code sequence to be used to encode its information signal. A receiver, knowing the code
10 sequences of the user, can decode the received signal to reproduce the original information signal. The use of the unique code sequence during modulation provides for an enlarging of the spectrum of the transmitted signal resulting in a spread spectrum signal. The spectral
15 spreading of the transmitted signal gives rise to the multiple access capability of CDMA.

If multiple users transmit spread spectrum signals at the same time, the receiver will still be able to distinguish a particular user's signal, provided that each user has a
20 unique code and the cross-correlation between codes is sufficiently low. Ideally, the cross-correlation should be zero, i.e., the codes should be orthogonal in the code space. Correlating a received signal with a code signal from a particular user will result in the despreading of
25 the information signal from that particular user, while signals from other users will remain spread out over the channel bandwidth.

However, the number of orthogonal codes in a system is limited. As a result, each cell has a limited number of
30 orthogonal channelization codes that are assigned different physical channels. The number of orthogonal channelization codes is dependent upon their spreading factor, which is related to the physical channel bitrates. This gives rise to the well-known downlink channelization code limitation
35 inherent in CDMA.

In radio resource management, congestion may occur e.g. on cell level. Several types of Radio Resource Management (RRM) functions exist, such as handover, power control, admission control and load control. The following are
5 examples from a radio system using Wideband Code Division Multiple Access (WCDMA), but similar things happens also in other mobile systems and might happen also in other systems.

The purpose of admission control is to ensure that there
10 are free radio resources for an intended call with required signal-to-interference ratio and bit rate or equivalent. The purpose of load control is to maintain the use of radio resources of the network within given limits.

Admission control is normally performed when a mobile
15 station initiates communications in a new cell, either through a new call or handover. Furthermore, admission control is performed when a new service is added during an active call. In general, the admission control procedure ensures that there exist a free code to use for a new call
20 and that the interference created after adding a new call does not exceed a prespecified threshold. Further, the admission control should also check that there is enough base station transmission power for the new call. Admission control should be done separately for uplink and downlink.
25 This is especially important if the traffic is highly asymmetric. Typical criteria for admission control are call blocking and call dropping. Call blocking occurs when a new user is denied access to the system. Call dropping means that a call of an existing user is terminated.

30 The basic principle of load control is the same as admission control. While admission control is carried out as a single event, load control is a continuous process where e.g. the interference is monitored. Load control measures the load factor of the cell, and, if the

predefined load factor is exceeded, i.e. the cell is congested, then the network may e.g. reduce the bit rate of certain users, delay the transmission for certain users or drop low priority calls. If there is an underload, load control may increase the bit rates of those users who can handle higher bit rates.

One version of load control is called channel switching (ChSw) or rate switching. The main idea is that if a user needs a low bit rate then he shares a common channel with other users. If the user should then need more capacity he can be switched over to a dedicated channel which is continuously reserved just for him. If the user on the other hand should need less capacity he can be switched to a common channel, if the user is using a dedicated channel.

A variant of channel switching is described in WO99/66748. Several methods are described on how to determine when to switch. According to one embodiment the buffer fill level can be used; if the queue length in the buffer is long this is a sign that more capacity is needed and vice versa. Two fixed thresholds in the buffer may be used to indicate when switching should take place, in order to create a hysteresis to avoid frequent switchings.

WO99/66748 further describes the case when there besides the previously mentioned buffer also exists a packet router buffer in a packet router. In this case it is described a second embodiment with a "back pressure" signal is transmitted from the buffer to the packet router if the queue length in the buffer becomes too long, whereupon temporarily buffering takes place in the packet router buffer rather than in the buffer.

In WO99/66748 these two embodiments can be combined and it can also be checked if the buffer is full because the connection is temporarily broken instead of because of

increasing traffic. Further, other traffic measures can be used, instead or as a complement, e.g. packet arrival time, packet arrival rate, packet density, connection's bit rate(s), current number of idle devices or current number
5 of idle spreading codes.

Thus, Radio Resource Management adapts the link bit rate after the load, while - given a certain link bit rate - Active Queue Management adapts the load to the link bit rate.

10 A problem is that the Radio Resource Management and Active Queue Management may have conflicts in the objectives for the buffer fill-level: Active Queue Management tries to maintain a 'low' buffer fill-level to improve interactivity over the link. A small buffer, on the other hand, makes it
15 difficult for Radio Resource Management to use the buffer fill-level as a measurement for prediction of future capacity needs of a link.

SUMMARY OF THE INVENTION

The object with the invention is to design a system with a
20 well behaving interplay between TCP congestion control, Active Queue Management and Radio Resource Management.

The problem with earlier solutions is that they have not realised that this has to be done. They have not understood that problems, such as oscillations, may occur in systems
25 where Active Queue Management and Radio Resource Management work independently. The interplay between Active Queue Management and TCP is in prior art fairly well understood. One assumption in prior art is, however, that the capacity of the bottleneck link remains constant. However, this is
30 not in line with the reality of a resource limited system.

In the invention it is noted that the Radio Resource Management and the Active Queue Management may have

conflicts in the objectives for the buffer fill-level: Active Queue Management tries to maintain a 'low' buffer fill-level to improve interactivity over the link. A small buffer, on the other hand, makes it difficult for Radio Resource Management to use the buffer fill-level as a measurement for prediction of future capacity needs of a link.

The solution according to the present invention is for upswitch:

- 10 - determining link resource status;
- if link congestion is determined then
 - (a)determining if it is possible to allocate more link capacity;
 - (b)allocating more link capacity when it is possible to
 - 15 allocate more link capacity;
 - (c)alleviating link congestion using Active Queue Management when it is not possible to allocate more link capacity.

An advantage with this method is that it can be ensured that Active Queue Management has not asked TCP to reduce its load at the same time as Radio Resource Management is providing more resources. The risk of conflicting actions is removed. As a consequence, the allocated capacity is better utilized, because the TCP load is not reduced prior to the up-switch of the capacity.

Further, the queue fill-state may not be a good measurement for Radio Resource Management, unless Active Queue Management and Radio Resource Management are integrated, as in the proposed method. With the present method, the main measurement for the Radio Resource Management decision is the up-switch request by Active Queue Management. Other measurements (like user activity statistics) may be used to support the Radio Resource Management decision.

The corresponding solution for forced downswitch will then be:

- determining cell resource status;
 - if cell congestion is detected then
- 5 (a) determining that it is necessary to switch down bit rate or rates in at least one link;
- (b) alleviating link congestion using Active Queue Management; and
- (c) switching down said bit rate or rates.

- 10 An advantage is because Active Queue Management is informed of the rate reduction in advance, it can start to reduce the source rate before the down-switch, thereby avoiding excessive buffering delays or buffer overflow.

- Further, because the link rate is still high when the
- 15 Active Queue Management actions start, the Active Queue Management actions (packet drop or ECN marking) take effect faster.

DESCRIPTION OF THE DRAWINGS

- Figure 1 discloses a schematic view of a system including
- 20 the invention
- Figure 2 discloses a method according to the most general aspect of the invention

DESCRIPTION OF PREFERRED EMBODIMENTS

- Figure 1 illustrates schematically, according to the
- 25 invention, a Universal Mobile Telecommunications system (UMTS) network 1 using Wideband Code Division Multiple Access (WCDMA). The UMTS network 1 includes a core network 2 and a UMTS Terrestrial Radio Access Network (UTRAN) 3. The UTRAN 3 includes a number of Radio Network Controllers
- 30 (RNC) 4 - of which only one is drawn for the sake of clarity. Each of the Radio Network Controllers 4 is coupled to a set of neighbouring base stations, normally called

Node Bs 5. Each Node B 5 is responsible for a given cell 11 and the controlling Radio Network Controller 4 is responsible for routing user and signalling data between that Node B 5 and the core network 2. A general outline of the UTRAN 3 is given in Technical Specification TS 25.401 V3.2.0 of the 3rd Generation Partnership Project. Figure 1 also illustrates mobile terminals or User Equipments (UE) 6, a Serving GPRS Support Node (SGSN) 7 and a GPRS Gateway Support Node (GGSN) 8. The Serving GPRS Support Node 7 and the Gateway GPRS Support Node 8 may e.g. provide packet switched data services to the User Equipment 6 via the UTRAN with the Gateway GPRS Support Node being coupled to e.g. the Internet 9, whereupon the User Equipment 6 may communicate with a node 10 connected to the Internet 9.

User data received at a Radio Network Controller 4 from the core network 2 is stored at a Radio Link Control (RLC) entity 12 in one or more buffers 13. User data generated at a User Equipment 6 is stored in buffers 14 of a peer Radio Link Control entity 15 at the User Equipment 6. User data (extracted from the buffers) and signalling data is carried between a Radio Network Controller 4 and a User Equipment 6 using Radio Bearers. Typically, a User Equipment is allocated one or more Radio Bearers each of which is capable of carrying a flow of user or signalling data. Radio Bearers are mapped onto respective logical channels. At a Media Access Control (MAC) layer, a set of logical channels is mapped in turn onto a transport channel. Several transport channels are in turn mapped at the physical layer onto one or more physical channels - which thus may include one or more links 19 - for transmission over the air interface between a Node B 5 and a User Equipment 6.

Each link is thus supported by one buffer in the Radio Network Controller 4 and one buffer in the User Equipment 6. Each of the buffers 13, 14 is controlled by Active Queue

Management (AQM) 16, 17 operating separately on each buffer 13, 14 to avoid link congestion. Further, in each Radio Network Controller 4 is included a Radio Resource Management 18, which controls the allocation of radio resources to channels and tries to avoid cell congestion.

According to other embodiments of the invention a buffer may work for more than one incoming and/or outgoing link. This may e.g. be the case in an Internet router. Further, Active Queue Management may work on more than one buffer simultaneously. In particular, an alternative would be to have a general Active Queue Management working to control the average traffic in a whole cell. Finally, of course, Active Queue Management needs not be performed on all buffers.

According to the present invention Radio Resource Management and Active Queue Management are co-ordinated. An overview of a process for upswitching is seen in Fig 2. First, link congestion is detected, block 21. Then it is determined if it is possible to allocate more link capacity, block 22. If it is possible to allocate more link capacity, then more link capacity is allocated, 23, else link congestion is alleviated using Active Queue Management.

According to one embodiment, link congestion may be detected by setting a congestion threshold Th in the buffer. When the queue length is longer than the congestion threshold Th , then link congestion is presumed to be near. The natural action would then be, according to prior art, to use Active Queue Management to take action. However, it might happen that the congestion is local and that it would be possible to allocate more bandwidth. Thus, according to the present invention, Radio Resource Management uses the same congestion threshold as an indication on a need for higher bit rate and determines if it is possible to

allocate more bandwidth. E.g. if the user using the congested link is using a common channel that he is sharing with other users, then it might be possible to instead allocate a dedicated channel just for him. Alternatively,
5 the user might e.g. be given a higher bit rate within the same common or dedicated channel.

Naturally, other criteria may be used alternatively to a congestion threshold or in combination therewith to take the decision to switch channel. Such criteria may be
10 traffic intensity, packet arrival times, time between packets etc.

When Radio Resource Management has determined if it is possible to allocate more bandwidth or not, this should be reported to Active Queue Management. This can be done by
15 signalling from Radio Resource Management to Active Queue Management. Alternatively, a timer can be introduced on the congestion threshold for the Active Queue Management. Thus, if the congestion threshold has been exceeded for a certain amount of time, then it can be presumed that Radio Resource
20 Management has no further bandwidth to allocate and that the Active Queue Management needs to take action.

If the Active Queue Management finds out that more bandwidth is allocated, it takes no action. However, if the Active Queue Management finds out that more bandwidth is
25 not allocated, then it may alleviate link congestion. This may be done by dropping or marking packets according to some predefined algorithm, preferably avoiding to drop subsequent packets by intent, in order to avoid causing TCP slow start in systems where TCP or similar is used. Marking
30 of packets may be done e.g. by setting an explicit congestion (ECN) flag in the header of a packet. When TCP or similar is used and the sender detects that the link is congested, then TCP will reduce its load and send data packets at a lower rate.

The congestion threshold may be fixed, but preferably it is movable. It can then be moved according to different primary criteria such as link characteristics i.e. e.g. round trip time (RTT) of the link and data rate or bit rate of the link. The link characteristics may then be used to calculate link capacity and thereby to set the congestion threshold. This gives a base value Th_{RRM} on the placement of the congestion threshold. It can be said e.g. that if the bit rate is high, then it can be permitted to have a longer queue length in the buffer and vice versa, considering that the buffer will be emptied quicker when using a higher bit rate.

In order to employ the Active Queue Management, according to one alternative, secondary criteria may also be used to move the congestion threshold, in order to give a more detailed placement of the congestion threshold. Then the congestion threshold Th may be calculated according to the following formula:

$$Th = Th_{RRM} + \Delta Th_{AQM} \quad (1)$$

where ΔTh_{AQM} is an adjustment calculated according to the secondary criteria according to Active Queue Management. This adjustment may e.g. be calculated by analysing whether a packet has been dropped or not. If a packet has been dropped, then ΔTh_{AQM} is increased with a positive constant δ , unless the congestion threshold Th has reached a maximum Active Queue Management threshold Th_{AQMmax} . If the queue length is determined to be less than the congestion threshold by a predetermined amount - preferably larger than δ in order to cause a hysteresis - then ΔTh_{AQM} is decreased with a positive constant δ unless the congestion threshold Th has reached a minimum Active Queue Management threshold Th_{AQMmin} . If none of these conditions are met, then ΔTh_{AQM} is left unchanged.

The maximum Active Queue Management threshold Th_{AQMmax} and the minimum Active Queue Management threshold Th_{AQMmin} , may be fixed or may be adjusted following the base value Th_{RRM} so that:

$$5 \quad Th_{AQM} = Th_{RRM} + a \quad (2)$$

$$Th_{AQM} = Th_{RRM} - b \quad (3)$$

where a and b are positive constants.

10 An alternative to analysing whether a packet has been dropped or not is e.g. to look at the buffering delay of each packet. Since the buffering delay is independent of bandwidth it may alternatively also be used as the only means for calculating the congestion threshold.

15 According to another alternative a maximum Active Queue Management threshold Th_{AQMmax} and a minimum Active Queue Management threshold Th_{AQMmin} are employed in a similar way as above, preferably adjusted following the base value Th_{RRM} , but using the base value Th_{RRM} as the congestion threshold Th :

$$Th = Th_{RRM} \quad (4)$$

20 Further, when Active Queue Management is to be employed, then for a queue fill level above the maximum Active Queue Management threshold Th_{AQMmax} incoming packets are discarded, for a queue fill level below the minimum Active Queue Management threshold Th_{AQMmin} incoming packets are not
25 discarded, and for a queue fill level between the maximum Active Queue Management threshold Th_{AQMmax} and the minimum Active Queue Management threshold Th_{AQMmin} incoming packets are discarded or marked with a certain probability.

30 An alternative to using a probabilistic approach when the queue fill level lies between the maximum Active Queue Management threshold Th_{AQMmax} and the minimum Active Queue

Management threshold Th_{AQMin} , is to use a counter to allow only one in every $(n+1)$ th packet to be dropped.

Other Active Queue Management algorithms may also be used or adapted in a similar way.

- 5 Downswitching may be done by Radio Resource Management primarily for two reasons. A first reason is that less capacity is needed because e.g. a user is needing less bandwidth or goes passive. A second reason is because of resource shortage in the cell due to e.g. many new users or
10 handovers, new services per user, users moving from the cell centre to the cell periphery (thus requiring a higher power, thus causing interference to others) etc

In the first case the capacity after the downswitch will normally be sufficient. If e.g. a user earlier had been
15 allocated a dedicated channel, a hysteresis threshold in the buffer may indicate a low usage. The same threshold as the congestion threshold may of course be used, but it is better to use a hysteresis threshold at a shorter queue length than the congestion threshold, to avoid unnecessary
20 frequent channel switchings. The hysteresis threshold may be fixed, but preferably it is on a fixed distance to the congestion threshold, thus moving when the congestion threshold is moving.

The user of a dedicated channel may now instead be switched
25 to a common channel, which may be sufficient for his needs. Naturally other criteria may be used alternatively to a hysteresis threshold or in combination therewith to take the decision to switch channel. Such criteria may be traffic intensity, packet arrival times, time between
30 packets etc.

In the second case with the forced downswitch, it is a cell congestion. A solution is then to e.g. switch to a lower bit rate for the user in a dedicated channel or for all or

some of the users in a common channel. Another solution is to switch the user in a dedicated channel into a common channel - and thus allocating him a lower bitrate. Yet other solutions are to delay the transmission for certain
5 users, to drop low priority calls etc.

Link congestion is, as a consequence, naturally very likely to occur and, depending on the actions taken, probably in more than one link simultaneously. Thus, in the second case the Radio Resource Management should inform the Active
10 Queue Managements for all links or for the affected links of the intention of a forced downswitch. This is preferably done by some type of signalling. Said Active Queue Managements can then take appropriate actions to avoid buffer overflow, such as dropping or marking packets.

15 For the sake of readability, we have in this disclosure explicitly referred to specific protocols, systems and functions. It should be clear, however, that the present invention is applicable to a broad range of systems, protocols and functions with similar properties as
20 described in this invention disclosure:

The present invention is applicable to any wireless system for packet-data transfer equipped with a resource management function, not only WCDMA. In fact the system need not even be wireless. However, considering that
25 wireless systems have the greatest problems with allocating resources, wireless systems will have the greatest advantages with the present invention

Further, the present invention is applicable independently of the choice of Active Queue Management algorithm.
30 Requirements for the present invention are a method for link congestion detection and a packet dropping/marking policy or other way of alleviating link congestion. A number of such Active Queue Management algorithms exist.

Further, the present invention is applicable to any type of packet-data traffic - not only using TCP - which traffic is equipped with an end-to-end load control mechanism. In particular, we note the ongoing efforts to make non-TCP
5 flows 'TCP-compliant' (TCP Friendly Rate Control, TFRC). The invention is also applicable to non-responsive flows, such as UDP. However, the congestion alleviation procedure in the link buffer may then follow a different pattern, in case the source rate is not reduced as a consequence of
10 packet losses.